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BASEBAND DEVELOPMENT SYSTEM IMPLEMENTATION

Syracuse University

Dr. Kamal Jabbour

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A baseband development system was successfully developed, installed and tested on a personal computer at Syracuse University. All pertinent technical information has been published in the open literature by Dr. Kamal Jabbour of Syracuse University.

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REAL-TIME SIMULATION OF VOICEBAND COMMUNICATION CHANNELS

J. F. Vega-Riveros, K. Jabbour and P. W. Dowd

Abstract- The real-time implementation of a voiceband communication channel simulator on the TMS32020 digital signal processor is presented. Several channel characteristics are simulated, including attenuation distortion, group delay distortion, harmonic distortion, listener echoes, frequency offset, gaussian noise, and impulsive noise.

1. INTRODUCTION

One of the problems encountered in evaluating the performance of various communications equipment- such as group delay meters [1], channel equalizers [2], and voiceband modems [3]- is the need to test them under realistic operating conditions, similar to those encountered during their actual use. These tests could be carried out on actual channels, but the cost and the need to access both ports of a channel make it prohibitive and impractical for all but a few equipment designers. Even then, using real lines gives the designer a limited choice of channel responses. Channel simulation then becomes not only a viable alternative, but a desirable one as well. (K?)

Channel simulators have traditionally been built with discrete analog components [3]. Active low-pass and band-pass filters, in some cases phase equalized, are used to simulate the amplitude response. Second-order all-pass filters simulate the group delay response, usually peaking in the middle of the pass-band. A

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cascade of low-pass, band-pass and all-pass filters may provide the designer with a dozen different channel responses. Noise may be injected into the system, and other channel impairments crudely approximated. The analog approach results in an inflexible design, with a very limited selection of simulated channels.

Digital computers have been used to simulate communication channels, usually offline, providing a fairly flexible solution. Such an approach suffered however from the absence of real-time simulation, limiting its suitability for equipment test and evaluation [4,5]. Recently however, large floating-point array processors such as the AP120B, and specially designed hardware, have offered a real-time digital approach [6]. The array processors and the dedicated hardware are usually driven by a host mainframe, on which network characterization and design are performed. The high cost of array processors and the lack of flexibility in dedicated hardware have limited their use in channel simulation.

In this document, we present a real-time digital channel simulator using Texas Instruments' TMS32020 single chip digital signal processor [7,8,9]. A sampling rate of 10 kHz is feasible, allowing the real-time simulation of voiceband channels with a bandwidth less than 4 KHz. An operating environment has been developed for a personal computer allowing channel characteristic parameter specification, design verification, TMS32020 code generation, code assembly, downloading object code, and TMS32020 execution initiation.

2. REAL TIME CHANNEL SIMULATION

The real time channel simulation is performed on the XDS/320, Texas Instruments' evaluation module for the TMS32020 digital signal processor, and its analog target board the AIB320. At a processor clock rate of 20 MHz, corresponding to an instruction cycle of 200 nsec, a sampling rate of 10 kHz is achieved, and is assumed in the following discussion. Twelve bit analog-to-digital (A/D) and

digital-to-analog (D/A) converters are used. A sixth-order lowpass Butterworth filter bandlimits the input signal to 4.7 KHz to minimize aliasing, and a similar filter smoothes the analog output.

The simulated channel characteristics can be divided into two groups: time-invariant and time-varying characteristics. The first group includes attenuation distortion, group delay distortion, harmonic distortion, listener echo, and frequency offset. The second group includes gaussian noise. and impulsive noise. In the following sections, we discuss the implementation of each of the above characteristics. A block diagram of the TMS32020 channel simulation is shown in Figure 1.

3. TIME-INVARIANT CHARACTERISTICS

3.1. Attenuation and Group Delay Distortions

Attenuation and group delay distortions are simulated with an FIR transversal filter. The user specifies the desired attenuation and group delay at up to 15 points between 300 and 4000Hz. The design program generates a 15th order best fit polynomial for the attenuation response, and another polynomial for the group delay response. The attenuation response is translated from decibels to a linear scale, and the group delay response is integrated with respect to frequency to give the phase response. Equally spaced samples of the amplitude and phase responses are used with an Inverse Fast Fourier Transform to give the desired impulse response.

The TMS32020 has 544 words of internal data memory, allowing the implementation of a 256-tap filter.

The frequency response of the analog filters at the input and output of the simulator (pre-A/D converter and post-D/A converter) has been measured, and

found to contribute a group delay distortion of about $30\mu s$, and an amplitude distortion of 2dB. Even more significant is the spectrum shaping of the form $H(\omega) = \frac{2sin(\pi\omega)}{\omega}$ caused by the D/A converter at high frequencies (2.4dB at 4KHz). Therefore the desired response is adjusted for this inherent distortion, so that the overall response of the simulator and the analog interface is as close as possible to the user specifications.

3.2. Harmonic Distortion

Harmonic distortion is commonly caused by iron saturation in 2-to-4 wire hybrid transformers used in telephone circuits to interface the local loop to toll trunks. We simulate harmonic distortion with a 5th order polynomial of the form

$$y(nT) = x(nT)+ax^2(nT)+bx^3(nT)+cx^4(nT)+dx^5(nT) \tag{1}$$

where $x(nT)$ is the sampled signal.

For a purely sinusoidal signal of the form $x(nT) = cos(n\omega T)$, the first five harmonics can be represented as:

$$cos(n\omega T) = cos(n\omega T) \tag{2.a}$$
$$cos(2n\omega T) = 2cos^2(n\omega T) - 1 \tag{2.b}$$
$$cos(3n\omega T) = 4cos^3(n\omega T) - 3cos(n\omega T) \tag{2.c}$$
$$cos(4n\omega T) = 8cos^4(n\omega T) - 8cos^2(n\omega T)+1 \tag{2.d}$$
$$cos(5n\omega T) = 16cos^5(n\omega T) - 20cos^3(n\omega T)+5cos(n\omega T) \tag{2.e}$$

The user specifies the percentage amount of distortion at each harmonic. Equations (2.a)-(2.e) are then used to calculate the coefficients of the 5th order polynomial in Equation 1, that give the desired harmonic distortion. If the signal is not a pure sinusoid, it can be represented as a sum of sinusoids since the signal is band limited. Harmonic distortion of a composite signal generates cross modulation components, as would happen with actual channels. It should be noted however, that no special precautions have been taken in the present implementation

against the aliasing effects of harmonic components larger than 5 KHz that may be generated by Equation 1.

3.3. Listener Echo

Echoes are caused by mismatches in the transmission medium, resulting in a reflection of a portion of the incident waveform. Thus echoes are time domain phenomena, and can be represented as:

$$y(nT) = x(nT) + ay(nT - mT) \qquad (3)$$

where $x(nT)$ is the incident signal, a is the attenuation and mT the delay of the echo path, and $y(nT)$ the resulting signal. Equation 3 has a z-transform with m poles representing the m complex roots of a. Therefore an echo can be expressed in the frequency domain and appears as a sinusoidal ripple in the frequency response, suggesting that echoes could be simulated with the same FIR filter used for the amplitude and group delay distortion. This approach fails however to simulate all the impairments that the echo picks in its trip through the channel, and results in a very long impulse response. We therefore chose to simulate listener echo by using a long first-in-first-out circular buffer. A delayed, scaled version of the output is digitally fed back to the input of the simulator implementing the return path of listener echo, the type of echo of interest in data transmission testing. A 2048 word circular buffer is used, allowing echoes with delays up to 204.8ms to be specified, in increments of $100\mu s$.

3.4. Frequency Offset

Frequency offset is simulated by single sideband modulation of the input signal with a very low frequency carrier, typically less than 5Hz (Figure 2). SSB modulation is efficiently performed by using the phase discrimination method, which uses two product modulators supplied with the input signal and its Hilbert

Transform, and carriers in phase quadrature to each other. In the case of a sinusoidal incoming signal x(nT), represented by $x(nT) = a\ cos(n\omega T)$ shifted by 90 degrees to obtain the quadrature component $v(nT) = -a\ sin(n\omega T)$ and then respectively multiplying each component by the sine and cosine of the carrier signal of frequency ω_m, and adding the products gives:

$$\begin{aligned} y(nT) &= a\ cos(n\omega T)cos(n\omega_m T) - a\ sin(n\omega T)sin(n\omega_m T) \\ &= a\ cos(n[\omega - \omega_m]T) \end{aligned}$$

This results in shifting the spectrum of the input signal by the value of the carrier frequency ω_m. [10]. Since an ideal 90 degree phase shifter (Hilbert Transformer) is non causal and cannot be implemented, an approximation of a 90 degree phase splitter consisting of a cascade of first order all pass filters has been used [11]. One arm of the splitter has two first order all pass filters with poles at r=0.737 and r=−0.00196, while the other arm has three first order all pass filters, with poles at r=0.924, r=0.439 and r=−0.586. The phase splitter has a unity amplitude response, and a fairly flat phase response at 90 degrees. The carrier frequency for the SSB modulator is generated by table look up from a 256-entry sinewave table. Carrier frequencies up to 32 Hz may be specified in steps of 0.01 Hz.

4. TIME-VARYING CHARACTERISTICS

Time-varying characteristics include Gaussian and impulsive noise. The simulation of the random characteristics use the pseudorandom number is based on [14]. The pseudorandom sequence is assumed uniformly distributed, and is used with Gaussian and Laplacian look up tables to obtain the respective distributions [12-14].

4.1. Gaussian Noise

Gaussian noise is generated by the thermal movement of electrons under the effect of an electric field. In our simulation, Gaussian noise is obtained from the uniformly distributed pseudorandom sequence, by using a 256-entry look-up table of the Normal (Gaussian) distribution. Some random noise - not necessarily Gaussian - is also generated during the digital implementation of the simulator from the effects of finite register length. Input signal quantization in the A/D converter, multiplier roundoff in the transversal filter, the first order all pass filters, and the harmonic distortion polynomial, and sample quantization in the sinewave look-up table all contribute some noise to the simulation. This noise is assumed Gaussian and is considered in the generation of the specified noise.

4.2. Impulsive Noise

Impulsive noise is caused by the switching currents of mechanical relays in the switched telephone network, and is the major source of catastrophic errors in data transmission. Several models have been suggested to simulate the effects of impulsive noise, e.g. bursts of errors in data transmission, but the literature is scarce in models for the actual noise. We have adopted a Laplacian distribution for modeling the amplitude of the impulses as suggested by Schwartz [15]. Impulsive noise is simulated from the pseudorandom number generator by using a look-up table of a Laplacian distribution, generated by the design program to meet the user-specified values of magnitude and frequency of occurrence of bursts.

5. THE CHANNEL SIMULATOR OPERATING ENVIRONMENT

Channel design is achieved through a personal computer by means of an interactive menu driven program written in Pascal. The operating environment controls all aspect of the XDS/320, including power up initialization and syn-

chronization. An operator may specify a new channel definition, or immediately download a previously defined simulation. The initial menu is shown in Figure 3.

The major component of the operating environment program are as follows: Model Specification, Model Verification, Source Construction, TMS32020 Code Assembly, Object Download and Execution.

5.1. Model Specification

The user specifies the parameters for those channel characteristics of interest during a given simulation. Figure 4 illustrates the format of the characteristics selection menu. For attenuation and group delay distortion, the user specifies the desired response at up to 15 frequencies. The residual response of the pre-A/D and the post-D/A filters prestored in the design program, and the spectrum shaping by the D/A calculated as a function of sampling rate, are taken into account in designing the channel. The specified response is fitted with a 15th order polynomial, from which the real and imaginary components of the frequency response are calculated. An IFFT is then applied, giving the desired impulse response. The user also specifies the desired number of harmonics and their magnitudes, the delay and attenuation of the return path for the listener echo, the frequency offset in Hz, the source frequency and amplitude of the phase jitter, the Gaussian noise in dBm, the threshold of impulsive noise and the number of bursts in a 15 minute interval, and the threshold and frequency of occurrence of gain hits.

5.2. Model Verification

The design program then allows the user to view the model of the designed channel, and compare it to the original specifications. Changes in the specifications can be made at this stage, before the generation of the TMS32020 code. The operator then decides whether to proceed with the verification and

assembly, start over, or exit.

5.3. Source Construction

This routine translates the parameters generated during model specification into a computer program in TMS32020 assembly language. It also assesses the feasibility of a real-time implementation of the model at the specified sampling frequency. Because of the limited power of the TMS32020 (500 instructions per sample at 10kHz) and the minimal amount of data memory (544 words), a tradeoff between the number of characteristics that can be simulated together and the tightness of these characteristics may be necessary. A worst case channel design with a 256-tap FIR filter and several other impairments requires a cascade of two or more TMS32020 modules for a real-time simulation.

5.4. TMS32020 Code Assembly

After the source code has been generated, the TMS32020 assembler supplied by Texas Instruments is invoked to produce the object code. The program is invoked by the channel simulator environment, and upon completion, it returns control to the environment.

5.5. Object Download

When the code assembly is completed, the object code is stored under a filename specified by the operator. The operator is then asked whether to download and execute the new channel definition. The program will comply with the operators order, then return to the main menu.

6. EXAMPLES

Several sample characteristics specifications and measurement results are shown in Figures 5-9 and Tables 1-3. Table 1 shows a sample attenuation distortion specification where 5 points where given in the range 300-4000 Hz, and Figure 5 shows the input/output characteristics of attenuation distortion obtained with the simulator.

Table 2 shows a delay distortion specified at 5 points in the range 300-4000Hz, and the results of measurements on the simulator. This delay distortion was measured by computing the derivative of the phase characteristic, as follows $\tau(f) = -\left\{\frac{\phi(f1) - \phi(f2)}{2\pi(f1 - f2)}\right\}$ where $\tau(f)$ is the group delay, $\phi(f1)$ and $\phi(f2)$ are the values of the phase characteristic at $f1$ and $f2$ respectively, and $f1$ and $f2$ are frequencies in the neighborhood of f, the frequency of interest.

Figure 6 shows the input and output of the channel simulator for a pure sinusoidal input when a 2.0 Hz frequency offset is specified. The output frequency was 2.08Hz above the input frequency.

Figure 7.a and 7.b show the effect of harmonic distortion as specified in Table 3. Figure 7.a shows the input signal spectrum and Figure 7.b the spectrum of the harmonically distorted signal.

In Figure 8 the power spectrum of the noise measured at the output of the channel simulator is shown. It can be observed that the spectrum is flat in the range of interest. The effect of the anti-aliasing filters can be observed on the right edge as a decreasing power density.

7. CONCLUSION

A versatile yet low cost telephone channel simulator has been designed. Several of the important channel characteristics have been simulated in real-time, providing a useful test tool for communications equipment. The simulator suffers

from the limited time available for real time processing, requiring multiple TMS32020 operating in parallel in order for all impairments to be simultaneously simulated.

8. ACKNOWLEDGEMENTS

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BIOGRAPHIES

Jose Fernando Vega-Riveros received the "Ingeniero Electronico" degree from the Pontificia Universidad Javeriana in Bogota, Colombia, in August 1979. He worked for the Communications Division of Avianca Airlines from 1979 to 1982, where he was involved in projects related to quality evaluation of telephone channels and data communication equipment, computer network architecture and communication management. He joined Syracuse University in 1982 where he obtained his MSEE in August 1983, and is currently working towards the PhD. His main research interests are in Data Communications, Computer Networks, and Digital Signal Processing

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Frequency (Hz)	Attenuation (dB)
300	5.0
1000	0.0
2000	0.0
3400	6.0
4000	9.0

Table 1: Sample Attenuation Distortion Specification

Frequency(Hz)	Group Delay(mSec)	
	Specification	Measurement
300	3.0	2.92
1000	1.5	1.57
2000	0.8	0.80
3400	2.0	1.99
4000	5.0	5.41

Table 2: Sample Group Delay Distortion. This results are relative to the delay introduced by the FIR filter.

Harmonic Component	Distortion (%)
2nd	10.0
3rd	5.0
4th	1.0
5th	0.0

Table 3: Sample Harmonic Distortion

Figure Captions

Figure 1: Block diagram of the TMS32020 channel simulation.

Figure 2: Frequency Offset.

Figure 3: Initial menu format.

Figure 4: Characteristic Selection Menu.

Figure 5: Attenuation distortion example.

Figure 6: Input/Output behavior of channel simulator for a pure sinusoidal input when a 2.0 Hz frequency offset is specified.

Figure 7: Harmonic distortion example. (a) Input spectrum. (b) Output spectrum.

Figure 8: Power spectrum of noise

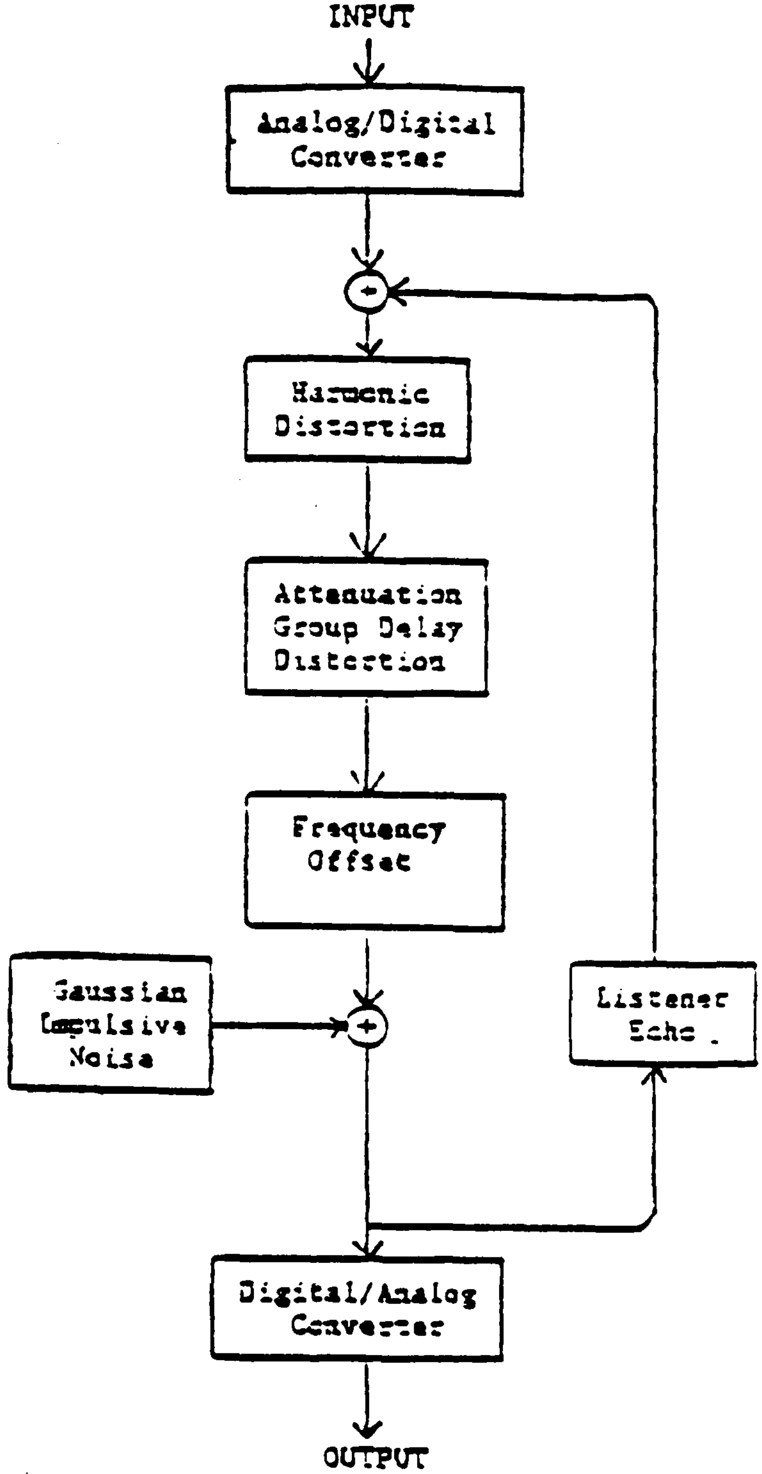

Figure 1. Block Diagram of Channel Simulator

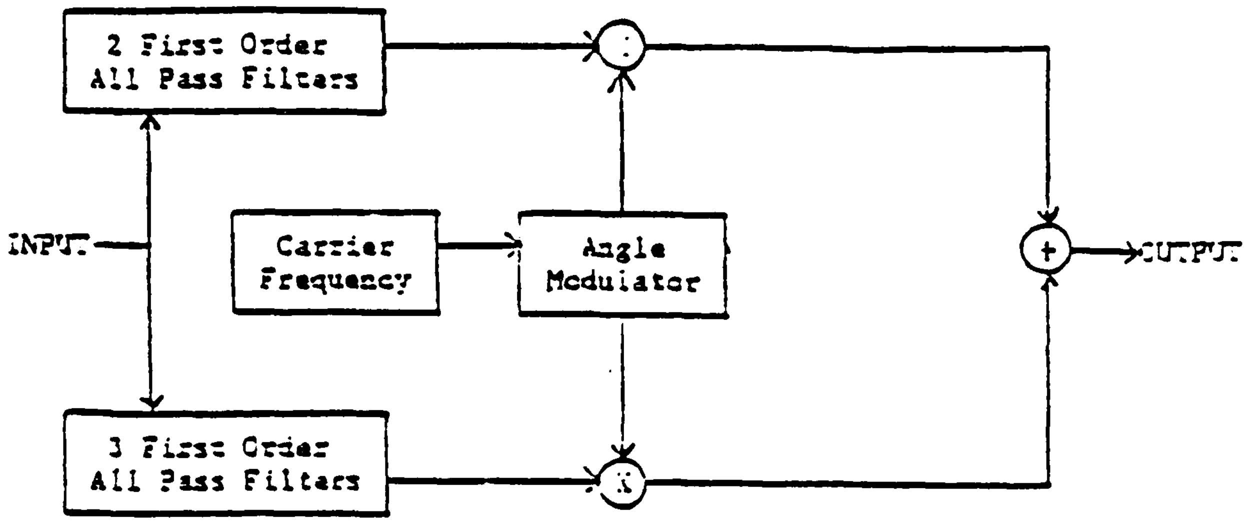

Figure 2. Frequency Offset

Telephone Channel Simulator

S : Specify new channel definition
D : Download and run an existing file
H : Help
Q : Quit

What is your selection?

Figure 3: Initial menu format.

Telephone Channel Simulator

A : Attenuation Distortion
D : Group Delay Distortion
H : Harmonic Distortion
G : Gaussian Noise
I : Impulsive Noise
E : Echo

V : Verify and Assembly
Q : Quit

What characteristic would you like to specify?

Figure 4: Characteristic Selection Menu.

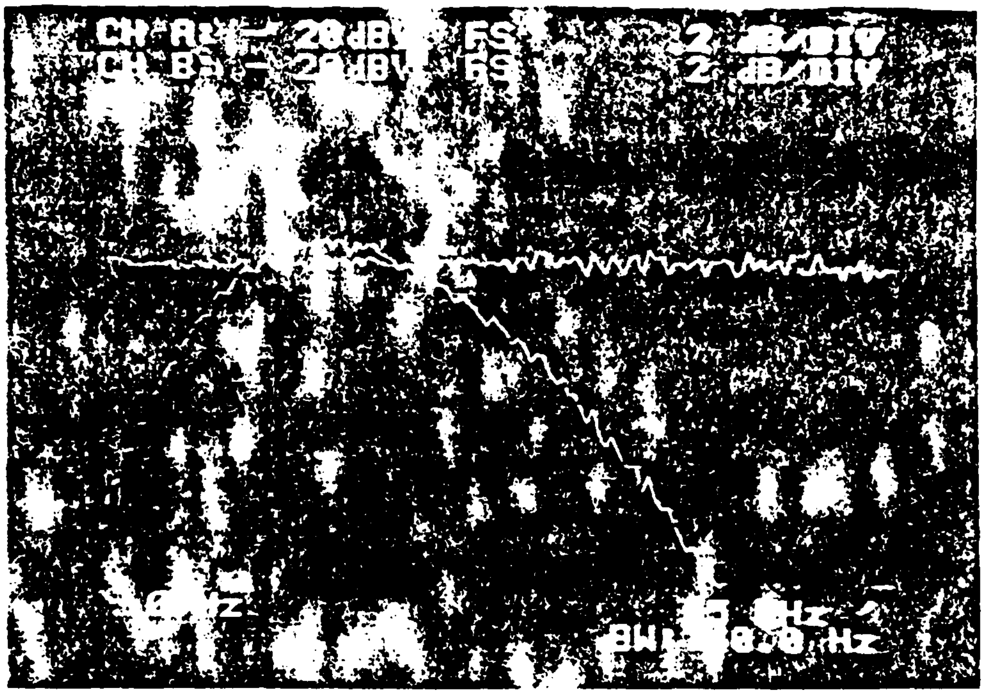

Fig. 5

Fig. 6

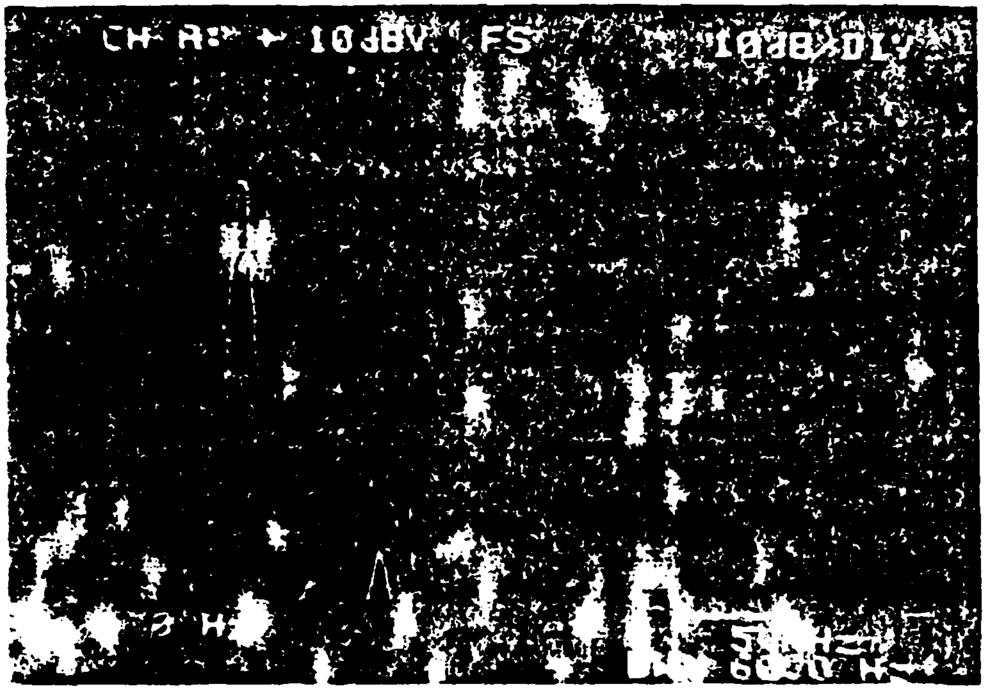

Fig. 7(a)

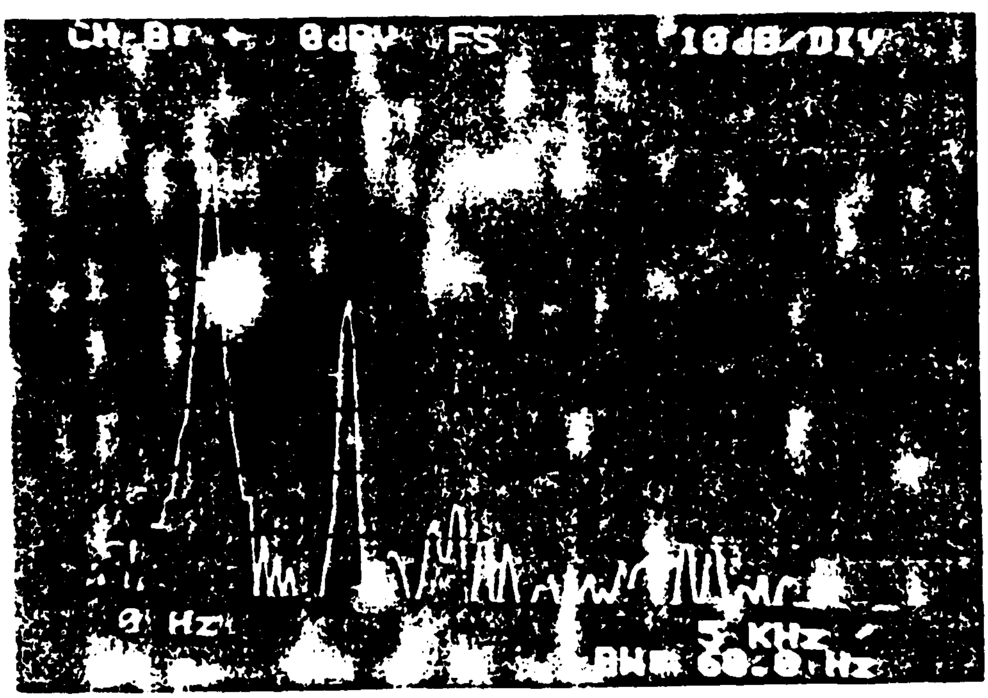

Fig. 7(b)

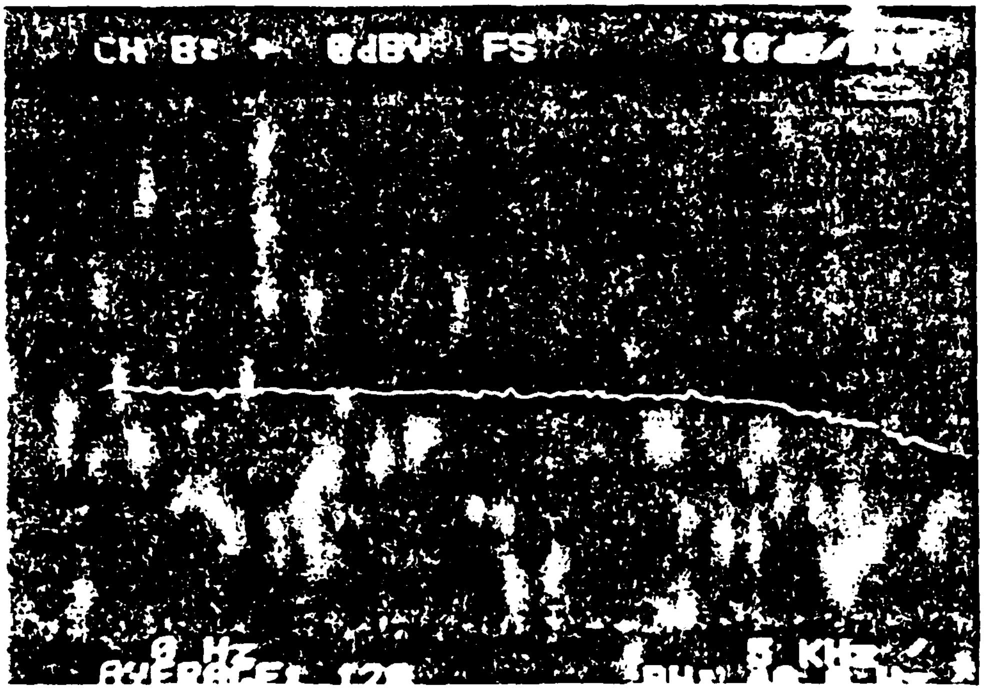

Fig. 8

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